## Remarks

The Applicants respectfully request reconsideration of the application in view of the following remarks.

In the Office action mailed July 14, 2006 ("Office action"), the Examiner rejected claims 1-7, 9-15, and 21-36 under 35 U.S.C. § 102(b) as being unpatentable over U.S. Patent No. 5,974,380 to Smyth ("Smyth"). The Examiner rejected claim 20 under 35 U.S.C. § 102(3) as being unpatentable over U.S. Patent No. 6,393,392 to Minde ("Minde"). Finally, the Examiner rejected claims 8, and 16-19 under 35 U.S.C. § 103(a) as being unpatentable over Smyth in view Minde. The Applicants respectfully disagree.

**Claims 1-7, and 9 Should Be Allowable**

Claim 1 recites: "measuring disparity between excitation patterns of *the individual input channels* of the multi-channel input audio signal [and] determining whether to encode the portion using joint channel coding or independent channel coding based at least in part on the *measured disparity between excitation patterns of the individual input channels.*" Smyth fails to teach or suggest the recited arrangement.

For example, the Office action asserts the following passages from Smyth:

> Hence, the pre-transient quantization noise is limited to a sub-subframe period.
> Transient Declaration
> A sub-subframe is declared transient if the ratio of its energy over the preceding sub-buffer exceeds a transient threshold (TT), and the energy in the preceding sub-subframe is below a pre-transient threshold (PTT).
>
> Col. 19, lines 14-17.

> Once the bit allocations (ABIT) and scale factors (SF) have been generated using the first estimation difference signal, their optimality may be tested by running a further ADPCM estimation process using the estimated ABIT and RMS (or PEAK) values in the ADPCM loop 72.
>
> Col. 16, lines 42-46.

> This is applicable at very high bit rates. The preferred approach is to retain the psychoacoustic bit allocation and allocate only the additional bits according to the mmse scheme.
>
> Col. 7, lines 26-28.

> As a result, the psychoacoustic allocation morphs to a mmse allocation as the bit rate increases thereby providing a smooth transition between the two techniques.
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>
> The encoder 12 employs a global bit management (GBM) system 30 that dynamically allocates the bits from a common bit-pool among the channels, between the subbands within a channel, and within an individual frame in a given subband.
>
> Col. 6, lines 65-67.

The indicated passages fail to teach or suggest "measuring disparity between excitation patterns of *the individual input channels*."

***Transient status is not disclosed in Smyth as a measured disparity between excitation patterns of individual input channels.***

For example, the Office action attributes "measuring disparity between excitation patterns of individual input channels" to the discussion of "transient status." Office action, page 2, ¶ 2. However, the above passage indicates that "the pre-transient quantization noise is limited to a sub-subframe period." Thus, Smyth limits "pre-transient quantization noise" by detecting and declaring transients. However, Smyth is declaring transients to limit pre-echo noise artifacts between consecutive sub-subframes within a subframe. See e.g., Smyth, col. 19, lines 14-17 and Figure 14a. This fails to teach or suggest; and teaches away from "measuring disparity *between excitation patterns of the individual input channels*."

For example, Smyth states that "if a rapid change in signal amplitude takes place in a block, i.e., a transient occurs ...." Col 18, lines 36-37. If any of the "sub-subframes are declared transient, two separate scale factors are generated for the analysis buffer, i.e. the current subframe." Col. 18, lines 64-67. This is not between channels. Further, Figure 14a of Smyth indicates that the transient detection is between sub-subframes of a subframe. Thus, Smyth is describing transient detection between sub-subframes. Col. 19, line 20, and lines 53-56. This teaches away from "measuring disparity between excitation patterns of *the individual input channels* of the multi-channel input audio signal [and] determining whether to encode the portion using joint channel coding or independent channel coding based at least in part on the *measured disparity between excitation patterns of the individual input channels.*."

Next, the Office action (page 2) directs Applicants to the following passages of Smyth:

> The GBM system 30 first selects the appropriate encoding strategy, which subbands will be encoded with the VQ and ADPCM algorithms and whether JFC will be enabled. Col. 24, lines 38-41.

> The GBM system 30 first decides which channels' subbands will be joint frequency coded and averages that data, and then determines which subbands will be encoded using VQ and subtracts those bits from the available bit rate. The decision of which subbands to VQ can be made a priori in that all subbands above a threshold frequency are VQ or can be made based on the psychoacoustic masking effects of the individual subbands in each frame. Thereafter, the GBM system 30 allocates bits (ABIT) using psychoacoustic masking on the remaining subbands to optimize the subjective quality of the decoded audio signal.
> Col. 7, lines 11-21.

> The encoder 12 employs a global bit management (GBM) system 30 that dynamically allocates the bits from a common bit-pool among the channels, between the subbands within a channel, and within an individual frame in a given subband.
> Col. 6, lines 65-67.

The above passages fail to teach or suggest "measuring disparity between excitation patterns of *the individual input channels* of the multi-channel input audio signal [and] determining whether to encode the portion using joint channel coding or independent channel coding based at least in part on the *measured disparity between excitation patterns of the individual input channels*." Smyth does describe a system that "decides which channels' subbands will be joint frequency coded and averages that data." Col. 11, lines 11-14. However, the provided language fails to teach or suggest "determining whether to encode ... using joint channel coding ... based ... on the measured disparity [between excitation patterns of the individual input channels]." There is no measured disparity between excitation patterns of individual input channels in Smyth. As quoted above, Smyth also describes dynamically allocating "the bits from a common bit-pool among the channels," but this also fails to teach or suggest "determining whether to encode ... using joint channel coding ... based ... on the measured disparity [between excitation patterns of the individual input channels]."

***Smyth describes employing joint frequency coding <u>at</u> low bit rates***

Smyth states that "[j]oint frequency coding may be employed at low bit rates to simultaneously encode multiple channels in the higher frequency subbands." Col. 6, lines 24-26. Thus Smyth describes employing joint frequency coding at low bit rates and teaches away from "determining whether to encode ... using joint channel coding ... based ... on the measured disparity [between excitation patterns of the individual input channels]."

Finally, Smyth states that the "encoder 12 may also use joint frequency coding techniques to take advantage of inter-channel correlations in the higher frequency subbands." Col. 6, line 67, through col. 7, line 3. However, inter-channel correlations in high frequency subbands are known to exist (e.g., Smyth, col. 14, line 7) and Smyth merely describes taking advantage of that known condition for *"**very low bit rate applications**"* (emphasis added, col. 14, line 2). Describing a "very low bit rate"--*teaches away.*

Thus, Smyth fails to teach or suggest "measuring disparity between excitation patterns of the individual input channels of the multi-channel input audio signal [and] determining whether to encode ... using joint channel coding or independent channel coding based ... on the measured disparity." For at least this reason, claim 1 is allowable. In view of the foregoing discussion, the Applicants will not belabor the merits of the separate patentability of dependent claims 2-7, and 9. Claims 1-7, and 9 should be allowable.

**Claim 8 Should be Allowable**

For the reasons stated above in claim 1, Smyth fails to teach or suggest "measuring disparity between excitation patterns of the individual input channels of the multi-channel input audio signal [and] determining whether to encode ... using joint channel coding or independent channel coding based ... on the measured disparity." Claim 8 depends from claim 1 and should be allowable over Smyth for at least this reason.

Additionally, the proposed Smyth-Minde combination also fails to teach or suggest "measuring disparity between excitation patterns of the individual input channels of the multi-channel input audio signal [and] determining whether to encode ... using joint channel coding or independent channel coding based ... on the measured disparity."

For example, Minde, at col. 4, lines 9-23, describes:

> The excitation analysis is performed to determine the best combination of fixed codebook vector (codebook index), gain $g_F$, adaptive codebook vector (lag) and gain $g_A$ that results in the synthetic signal vector $\{\hat{s}(n)\}$ that best matches speech signal vector $\{s(n)\}$ (here { } denotes a collection of samples forming a vector or frame). This is done in an exhaustive search that tests all possible combinations of these parameters (sub-optimal search schemes, in which some parameters are determined independently of the other parameters and then kept fixed during the search for the remaining parameters, are also possible). In order to test how close a synthetic vector $\{\hat{s}(n)\}$ is to the corresponding speech vector $\{s(n)\}$, the energy of the difference vector $\{e(n)\}$ (formed in an adder 26) may be calculated in an energy calculator 30.

One of ordinary skill in the art would understand that the above passage describes testing "how close a synthetic vector" compares to the "speech vector" it represents. Thus the passage measures the difference between a vector and a coded version of that vector. The proposed combination fails to teach or suggest "*measuring disparity between* excitation patterns of *the individual input channels.*" For at least this reason claim 8 is allowable.

**Claims 10-15 Should Be Allowable**

Claim 10 recites: "an excitation pattern disparity measuring component operative to produce a excitation pattern disparity measure of disparity in excitation patterns between channels."

Smyth fails to teach or suggest "an excitation pattern disparity measuring component operative to produce a excitation pattern disparity measure of disparity in excitation patterns between individual input channels." For at least this reason, claim 10 should be allowable. In view of the foregoing discussion, the Applicants will not belabor the merits of the separate patentability of dependent claims 11-15. Claims 10-15 should be allowable. Such action is respectfully requested.

**Claims 16-19 Should Be Allowable**

Claim 16 recites: "selecting … a portion of the transform domain coefficients for band truncation … and … suppressing the selected portion of the transform domain coefficients from

encoding in a compressed audio data stream." The proposed Smyth-Minde combination fails to teach or suggest the recited arrangement.

For example, the "time-to-frequency transform" described in Smyth (col. 23, lines 5-6) produces "a sequence of frequency coefficients" (col. 23, lines 24-25). Whereas, the SMR is used to determine "the number of bits" (col. 23, line 18) "required for quantization" (col. 23, line 21). If the "SMR < 0" (col. 23, line 19), then "no bits are required for quantization." Col. 23, line 21. Thus, Smyth as proposed in the Office action (page 14) fails to teach or suggest "band truncation" of a selected portion of "transform domain coefficients." The proposed passages simply indicates that no further "bits are required for quantization."

Minde also fails to teach or suggest "selecting ... a portion of the transform domain coefficients for band truncation ... and ... suppressing the selected portion of the transform domain coefficients from encoding in a compressed audio data stream." For example, the "open loop" (col. 10, line 66) in Minde operates on linear predicted codes (col. 10, line 65), and thus teaches away from "transform domain coefficients" and truncation thereof. Additionally, there is no suggestion of band truncation in Minde.

For at least these reasons, the proposed Smyth-Minde combination fails to teach or suggest claim 16. In view of the foregoing discussion, the Applicants will not belabor the merits of the separate patentability of dependent claims 17-19. Claims 16-19 should be allowable. Such action is respectfully requested.

**Claim 20 Should Be Allowable**

Claim 20 recites: "an open-loop band truncator operating to select a first selection of transform domain coefficients for band truncation." Minde fails to teach or suggest the recited arrangement.

For example, the Office action asserts the following passages from Minde:

> C. Determine (open loop) estimates of lags in LTP analysis (one set of estimates for entire frame or one set for smaller parts of frame, for example one set for each half frame or one set for each sub-frame).
>
> Col. 10, lines 66-67.
>
> The action of these three blocks may be expressed mathematically (in the time domain) as:
>
> $v(n)=g_A i(n-lag)=g_A d(lag) i(n)$

> where d denotes a time shift operator. Thus, excitation v(n) is a scaled (by $g_A$), delayed (by lag) version of innovation i(n). In the multi-channel case there are different delays $lag_{11}$, $lag_{22}$ for the individual components $i_1(n)$, $i_2(n)$ and there are also cross-connections of $i_1(n)$, $i_2(n)$ having separate delays $lag_{11}$, $lag_{22}$ for modeling inter-channel correlation. Furthermore, these four signals may have different gains $g_{A11}$, $g_{A22}$, $g_{A12}$, $g_{A21}$.
> Col. 7, lines 24-34.

Applicants respectfully assert that the above passages fail to teach or suggest "band truncator operating to select a first selection of transform domain coefficients for band truncation." As described in Minde, a "lag" is a delayed version of a signal. Col. 3, line 49. An adaptive codebook contains "past excitations i(n) that are shifted into the codebook." Col. 3, lines 54-55. Finally, the gain operator is not disclosed as a band truncator. Thus, the above passages from Minde fails to teach or suggest the recited language.

For at least this reason, claim 20 should be allowable.

**Claims 21-25 Should Be Allowable**

Claim 21 recites: "selectively suppressing at least one of the joint coding channels as a function of at least quality of reproduction, rate control buffer fullness, and channel separation." Smyth fails to teach or suggest the recited arrangement.

For example, the Office action asserts the following passages from Smyth:

> The high frequency subband samples as well as the predictor coefficients are encoded using vector quantization (VQ). The VQ start subband can be fixed or may vary dynamically as a function of signal characteristics. VQ works by allocating codes for a group, or vector, of input samples, rather than operating on the individual samples. According to Shannon's theory, better performance/bit-rate ratios can always be obtain by coding in vectors. Col. 12, lines 55-63.

> As shown in FIG. 6, the frame grabber 64 shown in FIG. 5 varies the size of the window 79 as the transmission rate changes for a given sampling rate so that the number of bytes per output frame 80 is constrained to lie between, for example, 5.3 k bytes and 8 k bytes. Tables 1 and 2 are design tables that allow a designer to select the optimum window size and decoder buffer size (frame size), respectively, for a given sampling rate and transmission rate. At low transmission rates the frame size can be relatively large. This allows the encoder to exploit the non-flat variance distribution of the audio signal over time and improve the audio coder's performance. For example, at a sampling rate of 48 kHz and a transmission rate of 384 kbps the optimum frame size is 4096 samples,

> which is split into 4 subframes of 1024 samples. At high rates, the frame size is reduced so that the total number of bytes does not overflow the decoder buffer. Col. 10, lines 1-17.

The above passage from Smyth fails to teach or suggest "selectively suppressing at least one of the joint coding channels as a function of at least quality of reproduction, rate control buffer fullness, and channel separation."

For example, the first passage states that "VQ works by allocating codes for a group, or vector, of input samples, rather than operating on the individual samples." This fails to directly or inherently disclose the recited arrangement. No channel is suggested as suppressed for any reason. Allocating codes for a group or a vector of input samples rather than operating on individual samples merely discloses vector quantization. The second passage states that the "frame grabber … varies the size of the window … as the transmission rate changes for a given sampling rate so that the number of bytes per output frame 80 is constrained to lie between, for example, 5.3 k bytes and 8 k bytes," for example, "so that the total number of bytes does not overflow the decoder buffer." Grabbing frames to vary a window size based on transmission rate also fails to teach or suggest the recited arrangement.

Thus, the above passages from Smyth fail to teach or suggest the recited language. For at least this reason, claim 21 is allowable. In view of the foregoing discussion, the Applicants will not belabor the merits of the separate patentability of dependent claims 22-25. Claims 21-25 should be allowable.

**Claims 26-27 Should Be Allowable**

Claim 26 recites: "a channel suppressor operative to selectively suppress at least one of the joint channels based on at least one suppression parameter, wherein the suppression parameters comprise values of a current quality of audio reproduction, a current rate buffer fullness, and a current channel separation." Smyth fails to teach or suggest the recited arrangement.

For example, the Office action asserts the following passages from Smyth:

> The high frequency subband samples as well as the predictor coefficients are encoded using vector quantization (VQ). The VQ start subband can be fixed or may vary dynamically as a function of signal characteristics. VQ works by allocating codes for a group, or vector, of input samples, rather

> than operating on the individual samples. According to Shannon's theory, better performance/bit-rate ratios can always be obtain by coding in vectors. Col. 12, lines 55-63.

> As shown in FIG. 6, the frame grabber 64 shown in FIG. 5 varies the size of the window 79 as the transmission rate changes for a given sampling rate so that the number of bytes per output frame 80 is constrained to lie between, for example, 5.3 k bytes and 8 k bytes. Tables 1 and 2 are design tables that allow a designer to select the optimum window size and decoder buffer size (frame size), respectively, for a given sampling rate and transmission rate. At low transmission rates the frame size can be relatively large. This allows the encoder to exploit the non-flat variance distribution of the audio signal over time and improve the audio coder's performance. For example, at a sampling rate of 48 kHz and a transmission rate of 384 kbps the optimum frame size is 4096 samples, which is split into 4 subframes of 1024 samples. At high rates, the frame size is reduced so that the total number of bytes does not overflow the decoder buffer. Col. 10, lines 1-17.

The above passage from Smyth fails to teach or suggest "a channel suppressor operative to selectively suppress at least one of the joint channels based on at least one suppression parameter, wherein the suppression parameters comprise values of a current quality of audio reproduction, a current rate buffer fullness, and a current channel separation."

For example, the first passage states that "VQ works by allocating codes for a group, or vector, of input samples, rather than operating on the individual samples." This fails to directly or inherently disclose the recited arrangement. No channel is suggested as suppressed for any reason. Allocating codes for a group or a vector of input samples rather than operating on individual samples merely discloses vector quantization. The second passage states that the "frame grabber … varies the size of the window … as the transmission rate changes for a given sampling rate so that the number of bytes per output frame 80 is constrained to lie between, for example, 5.3 k bytes and 8 k bytes," for example, "so that the total number of bytes does not overflow the decoder buffer." Grabbing frames to vary a window size based on transmission rate also fails to teach or suggest the recited arrangement. Again, no channel is suggested as suppressed for any reason.

Thus, the above passages from Smyth fail to teach or suggest the recited language. For at least this reason, claim 26 is allowable. In view of the foregoing discussion, the Applicants will

not belabor the merits of the separate patentability of dependent claim 27. Claims 26-27 should be allowable.

**Claims 28-34 Should Be Allowable**

Claim 28 recites: "encoding the quantization step-size values of the quantization bands in the quantization matrix." Smyth fails to teach or suggest the recited arrangement.

For example, the Office action asserts the following passage from Smyth:

> The high frequency subband samples as well as the predictor coefficients are encoded using vector quantization (VQ). The VQ start subband can be fixed or may vary dynamically as a function of signal characteristics. VQ works by allocating codes for a group, or vector, of input samples, rather than operating on the individual samples. According to Shannon's theory, better performance/bit-rate ratios can always be obtain by coding in vectors. Col. 12, lines 55-60.

The above passage from Smyth fails to teach or suggest "encoding the quantization step-size values of the quantization bands in the quantization matrix."

For example, the passage states that "VQ works by allocating codes for a group, or vector, of input samples, rather than operating on the individual samples." This fails to directly or inherently disclose the recited arrangement. No matrix encoding of step-size values of quantization bands is suggested. Allocating codes for a group or a vector of input samples rather than operating on individual samples merely discloses vector quantization.

Further, Smyth states:

> Table 9 is a fixed down matrix table for an 8-ch decoded audio signal.
> Col. 5, lines 35

However, the described matrix provides down matrixing so that "decoded audio channels can then be redirected to match the physical output channel arrangement on the decoder hardware." Col 48, lines 48-50. Smyth further describes channel down matrixing as:

> The down matrixing equations for 5-channel source audio to a two-channel Lt Rt playback system are given by:
> Left=left+0.7*center-0.7*(lt surround+rt surround)
> Right=right+0.7*center+0.7*(lt surround+rt surround) Col. 50, lines 42-45.

Thus, Smyth's discussion of a matrix fails to teach or suggest "encoding the quantization step-size values of the quantization bands in the quantization matrix." For at least this reason,

claim 28 is allowable. In view of the foregoing discussion, the Applicants will not belabor the merits of the separate patentability of dependent claims 29-34. Claims 28-34 should be allowable.

**Claims 35-36 Should Be Allowable**

Claim 35 recites: "a quantization matrix encoder for encoding a quantization matrix in a header for a frame of the input audio signal, the encoding comprising encoding the quantization step-sizes of the quantization bands in the quantization matrix, the quantization matrix encoder further operating to identify any quantization bands with zeroed transform coefficients." Smyth fails to teach or suggest the recited arrangement.

For example, the Office action asserts the following passages from Smyth:

> The controller 106 calculates the transient modes (TMODE) for each subframe in each subband. The TMODEs indicate the number of scale factors and the samples in the estimated difference signal ed(n) buffer when PMODE=1 or in the input subband signal x(n) buffer when PMODE=0, for which they are valid. Col. 18, lines 23-27.
>
> The multiplexer 32 shown in FIG. 12 packs the data for each channel and then multiplexes the packed data for each channel into an output frame to form the data stream 16. The method of packing and multiplexing the data, i.e. the frame format 186 shown in FIG. 25, was designed so that the audio coder can be used over a wide range of applications and can be expanded to higher sampling frequencies, the amount of data in each frame is constrained, playback can be initiated on each sub-subframe independently to reduce latency, and decoding errors are reduced.
> Col. 31, lines 41-48.

The above passage from Smyth fails to teach or suggest "a quantization matrix encoder for encoding a quantization matrix in a header for a frame of the input audio signal, the encoding comprising encoding the quantization step-sizes of the quantization bands in the quantization matrix, the quantization matrix encoder further operating to identify any quantization bands with zeroed transform coefficients."

The passage does not describe "a quantization matrix encoder for encoding a quantization matrix in a header." Smyth describes only a channel down matrix. Col. 50, lines 42-45. The passage does not describe "encoding a quantization matrix in a header for a frame of the input

audio signal, the encoding comprising encoding the quantization step-sizes of the quantization bands in the quantization matrix." Again, only a channel down matrix is described.

Thus, Smyth's discussion of a matrix fails to teach or suggest "a quantization matrix encoder for encoding a quantization matrix in a header for a frame of the input audio signal, the encoding comprising encoding the quantization step-sizes of the quantization bands in the quantization matrix, the quantization matrix encoder further operating to identify any quantization bands with zeroed transform coefficients." For at least this reason, claim 35 is allowable. In view of the foregoing discussion, the Applicants will not belabor the merits of the separate patentability of dependent claim 36. Claims 35-36 should be allowable.

**Interview Summary**

On November 7, 2006, the Applicant's attorney, Daniel H. Bell, and the United States Patent and Trademark Office's Examiner, Myriam Pierre, discussed the Office action mailed July 14, 2006 during a teleconference. The attached Exhibit, "Talking Points for Examiner Interview scheduled November 7, 2006, 2pm, Application 10/016,918" was provided to the Examiner on October 30, 2007, via e-mail. Applicants representative discussed why they believe Smyth fails to teach or suggest "measuring disparity between excitation patterns of individual channels of the multi-channel input audio signal ... and determining whether to encode ... using joint channel coding or independent channel coding based ... on the measured disparity" as described in the Exhibit. No agreement was reached during the teleconference.

## CONCLUSION

Claims 1-36 should be allowable. Such action is respectfully requested. The Examiner is invited to call the undersigned attorney at the telephone number below if the Examiner believes that doing so would further the prosecution of the present application.

Respectfully submitted,

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By ______________________
Daniel H. Bell
Registration No. 56,141

**Talking Points for Examiner Interview scheduled November 7, 2006, 2pm**

**Application 10/016,918**

**Claims 1-7, and 9 Should Be Allowable**

Claim 1 recites: "measuring disparity between excitation patterns of individual channels of the multi-channel input audio signal … and determining whether to encode … using joint channel coding or independent channel coding based … on the measured disparity." Smyth fails to teach or suggest the recited arrangement.

For example, the Office action asserts the following passages from Smyth:

> Hence, the pre-transient quantization noise is limited to a sub-subframe period.
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> A sub-subframe is declared transient if the ratio of its energy over the preceding sub-buffer exceeds a transient threshold (TT), and the energy in the preceding sub-subframe is below a pre-transient threshold (PTT).
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> the channels, between the subbands within a channel, and within an individual frame in a given subband.
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The indicated passages fail to teach or suggest "measuring disparity between excitation patterns of individual channels of the multi-channel input audio signal."

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For example, Smyth states that "if a rapid change in signal amplitude takes place in a block, i.e., a transient occurs ...." Col 18, lines 36-37. If any of the "sub-subframes are declared transient, two separate scale factors are generated for the analysis buffer, i.e. the current subframe." Col. 18, lines 64-67. Further, Figure 14a of Smyth indicates that the transient detection is between sub-subframes of a subframe. Thus, Smyth is describing transient detection between sub-subframes. Col. 19, line 20, and lines 53-56. Transient detection ***between consecutive sub-subframes*** fails to teach or suggest the recited arrangement. This teaches away from "measuring disparity between excitation patterns of individual channels of the multi-channel input audio signal [and] determining whether to encode ... using joint channel coding or independent channel coding based ... on the measured disparity."

Next, the Office action (page 2) directs Applicants to the following passages of Smyth:

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> The GBM system 30 first decides which channels' subbands will be joint frequency coded and averages that data, and

> then determines which subbands will be encoded using VQ and subtracts those bits from the available bit rate. The decision of which subbands to VQ can be made a priori in that all subbands above a threshold frequency are VQ or can be made based on the psychoacoustic masking effects of the individual subbands in each frame. Thereafter, the GBM system 30 allocates bits (ABIT) using psychoacoustic masking on the remaining subbands to optimize the subjective quality of the decoded audio signal.
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> the channels, between the subbands within a channel, and within an individual frame in a given subband.
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> 65-67.

The above passages fail to teach or suggest "measuring disparity between excitation patterns of individual channels of the multi-channel input audio signal [and] determining whether to encode ... using joint channel coding or independent channel coding based ... on the measured disparity." Smyth does describe a system that "decides which channels' subbands will be joint frequency coded and averages that data." Col. 11, lines 11-14. However, the provided language fails to teach or suggest "determining whether to encode ... using joint channel coding ... based ... on the measured disparity [between excitation patterns of individual channels]." As quoted above, Smyth also describes dynamically allocating "the bits from a common bit-pool among the channels," but this also fails to teach or suggest "determining whether to encode ... using joint channel coding ... based ... on the measured disparity [between excitation patterns of individual channels]."

***Smyth describes employing joint frequency coding <u>at</u> low bit rates***

Smyth states that "[j]oint frequency coding may be employed at low bit rates to simultaneously encode multiple channels in the higher frequency subbands." Col. 6, lines 24-26. Thus Smyth describes employing joint frequency coding at low bit rates and teaches away from "determining whether to encode ... using joint channel coding ... based ... on the measured disparity [between excitation patterns of individual channels]."

Finally, Smyth states that the "encoder 12 may also use joint frequency coding techniques to take advantage of inter-channel correlations in the higher frequency subbands." Col. 6, line 67, through col. 7, line 3. However, inter-channel correlations in high frequency subbands are known to exist (Smyth, col. 14, line 7) and Smyth merely describes taking advantage of that known condition for ***"very low bit rate applications"*** (emphasis added, col. 14, line 2).

Thus, Smyth fails to teach or suggest "measuring disparity between excitation patterns of individual channels of the multi-channel input audio signal [and] determining whether to encode ... using joint channel coding or independent channel coding based ... on the measured disparity." For at least this reason, claim 1 is allowable. In view of the foregoing discussion, the Applicants will not belabor the merits of the separate patentability of dependent claims 2-7, and 9. Claims 1-7, and 9 should be allowable.